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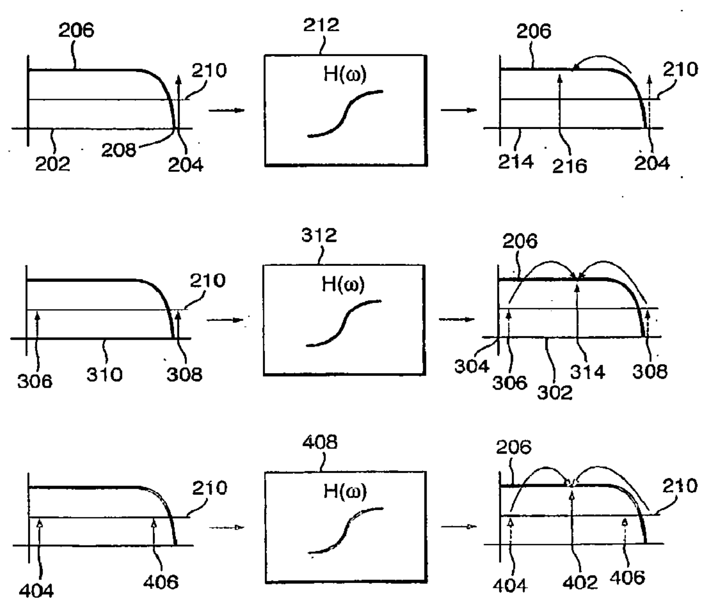
(54) Title: DIGITAL CONTENT PROTECTION



(57) **Abstract:** The present invention relates to digital content protection. Embodiments of the present invention provide, for example, a CD having data representing conventional audio or from which such audio can be derived using an intended target system together with data representing a spoiling component from which spoiling noise is created when the data is processed by a system other than the target system.

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DIGITAL CONTENT PROTECTION

Field of the Invention-

The present invention relates to digital content protection for digital content such as, for example, content stored on CDs, DVDs, digital tapes, or in digital data files and the like.

Background to the Invention

There is currently a world-wide problem relating to counterfeit audio-visual consumables, such as, for example, audio and video CDs, digital tapes (e.g. DAT, DCC etc.), DVSs, audio files, such as .WAV files and .MP3 files and the like. These media, or more accurately, the data stored on such media, or contained within such files, can often be digitally copied using a computer and, optionally, a CD/DVD writer and/or a digital tape recorder without suffering any loss in quality of the audio or video derived from such data. Even sophisticated anti-copying techniques can often be avoided using relatively unsophisticated programming techniques.

Furthermore, and as outlined below, a protected digital medium or protected digital content, when converted into an analogue format e.g. an analogue audio output, is generally unprotected and may easily be copied by analogue-to-analogue or analogue-to-digital means such as a microphone and a PC soundcard.

Many techniques are known in the art for preventing digital-to-digital copying of media or data, especially audio data such as, for example, music. However, in most cases audio data stored digitally using an appropriate data carrier must be converted ultimately into physical sound waves by an appropriate digital-to-analogue device for amplification and to drive conventional loudspeakers. These digital-to-analogue devices include, for example, PC soundcards and dedicated digital-to-analogue electronic circuitry in CD and DVD players etc. In the interests of quality, the sound/analogue signal produced by such devices is created as close as possible to the original medium base band recording so as to represent a reasonable facsimile of

the original data. Therefore this data (generally audio data) is then prone to being recorded or captured by an analogue device or an analogue-to-digital device such as, for example, a tape recorder or PC soundcard or the like. It would be advantageous to prevent, or at least reduce or dissuade, such copying to safeguard revenue streams for owners of recorded audio or other copyright material. This also applies to any other medium such as DVDs and the like.

It is an object of the embodiments of the present invention at least to mitigate some of the problems of the prior art.

Summary of Invention

Accordingly, a first aspect of the present invention provides a method of creating a digital signal representing an audio signal that is resistant to copying, the method comprising:

receiving a source digital signal having a predetermined sampling frequency;

generating a spoiling noise signal having a centre frequency below half said predetermined sampling frequency, said spoiling noise signal being imperceptible to a human with normal hearing if reproduced faithfully and having a frequency and amplitude such that when transformed with a non-linear transformation a human-perceptible noise is created; and

generating an output digital signal that is resistant to copying by combining said source digital signal and said spoiling noise signal.

If data representing an audible or base band signal is copied, that is, re-recorded, the non-linearities of the recording system are exploited to move energy at predetermined frequencies to create spoiling noise. The spoiling noise is arranged to fall within the audible frequency range at a level that is perceivable by humans.

The present invention provides spoiling noise components that are transferred by any non-linearities of a playback or recording device used for re-recording, especially any analogue stage in that process, to create unacceptable noise. The non-linearities are preferably characteristics of components of the device, e.g. amplifiers or transducers, but effects with other noise signals generated by the device may also

be exploited. As a further hindrance to rerecording, noise components that interact with commonly used tape bias frequencies may be employed.

A particularly advantageous form of noise signal comprises an ultrasonic or non-audible carrier frequency and modulated by a modulation envelope forming substantially symmetric packets. The packets have a size and shape determined so that a component of the reproducing or rerecording device breaks the symmetry of the packet and creates noise in the audible region. The component may be an automatic level or gain control or a feature of a digital coding scheme that aims to reduce the data volume.

This bias noise may also be encoded in such a way as to introduce packets of ultrasound and/or near-audible sound. That is, ultrasound tones (or near audible tones) in packets of calculated shape and duration may be placed amongst the bias and ultrasound/near audible balanced spike noise. If the packets are of correct geometry, then no or very low or practically imperceptible audible noise will be added due to their presence. However, the packet durations may represent a frequency that is easily audible (e.g. 2000 Hz). In this way, although the ultrasound (or near audible) tones produce no or very little audible component during playback, their modulation (packet geometry) may affect the automatic level control of a recording device therefore modulating the recording level of the copy at an audible frequency. It is to be noted that automatic level controls are generally characterised by a fast attack, slow relax profile. Therefore, a detected ultrasound/near audible sound packet of appropriate duration (e.g. 70 Hz to 2000 Hz) will add a noise component to the recording at or near these frequencies that may be heard when played back. The packet repetition rate may also contribute to this process. To prevent or suitably reduce noise created by the sudden appearance of these packets, suitable packet geometries may be devised (e.g. a rampdown, rampup, rampdown profile). In this way, the rate of level change may be controlled so as to reduce any possible audible spike noise and the rise/fall rate ratio may be controlled for the purpose of optimising noise when psychoacoustic compression is encountered, e.g. MP3 (M-PEG layer 3 compression) where sudden sound peaks are presumed to be

momentarily dominant therefore allowing less powerful sound beneath the peak and components during the 20ms ear relaxation period to be removed without affecting the perceived sound quality. By controlling the rise/fall rate it is therefore possible to force the MP3 codec to remove part of the falling ramp so as to break the symmetry of the noise packet and thereby create further audible noise. Furthermore, the packet duration may be devised so that it corresponds to a frequency component within the important or preferred psycho-acoustic range (e.g. 2 kHz) for the codec, or beyond that region should it be necessary to include or exclude the noise packet for noise component or data encoding purposes.

The exact geometry and rise/stroke fall ratios of the packets depend on the maximum amplitude level for a given frequency that is perceptible according to the psycho-acoustic model of human sound perception, but in general the ramp rate ratio, that is the ramp rate divided by the packet width, should preferably be in the range of 1/3 to 1/10 and optimally in the range of 1/3 to 1/5 for best effect with automatic level control circuits. The packet width is preferably in the range of 100 samples to 10,000 samples (at CD rate) with a 1 to 5 ramp rate ratio preferred at the lower end of that range. Thus, the ramp period is preferably of the order of 1 millisecond, which is within the Haan window of ear basillar membrane reflex period, typically 5 milliseconds. For maximum effect with coding schemes that make use of psycho-acoustic effects, the packet length after the end of ramp up period should be equal to the Haan window as this is the period that will be discarded by the codec producing the greatest asymmetry in the coded packet. For automatic level or gain control circuits with slower responses, the packet length should be longer. Multiple packets with different lengths, geometries and spacings may be used.

Preferred embodiments provide a method further comprising the step of encoding data within or using at least some of the noise packets. Advantageously, the encoded data can be used for a number of purposes such as, for example, identifying the original of the base band signal.

Preferably, embodiments provide a method in which the data provides an indication of at least one of information associated with the digital content,

information associated with a publisher of the digital content, information associated with a purchaser of the digital content.

More preferably, embodiments provide a method in which the data provides an indication of at least the composer of a digital content, a track name associated with the digital content.

Such noise packets may be introduced in a prescribed manner so as to represent Pulse Interval Coded information. For example, the presence (or absence) of a packet among the other encoded noise may be detected by either recognition of its known parameter range (frequency of noise contained, ramp rate, profile, duration, level limits etc.) or the change of encoded noise parameters (absence/presence of frequency/frequencies noise level, sound level etc.) such as may be routinely performed by realtime frequency analysis using a frequency/spectrum analyser or Fourier-based program performing that function. If for example, the range between packets was of a 1 to 16 ratio, then after recognition of the synchronisation/start point, say by expecting five identical packet spacings, then the spacing between each subsequent packet would represent a value of 1 to 16 (or 0 to 15) so as to represent one half of one byte of information. In other words, two subsequent packets would represent one byte of information which could in turn represent one ASCII or Unicode character. In turn, several of these characters could make a repeated character string such as serial number, purchaser's name etc. In this way, a short excerpt could be analysed so as to retrieve the encoded information/character string even if the packets have been degraded (or removed) by compression codes.

Viewed another way, noise may be introduced by carefully prescribed modulation geometry of various noise components so that any noise reproduced in the relevant playback system is below the threshold of human hearing or so that noise of any frequency that is added in any other way may be reproduced by the playback system (before copying) at a level that is below the threshold of human hearing.

For the purposes of the present application, ultrasonic noise is defined as noise having a frequency just beyond an upper threshold of the human range of hearing. In general, this threshold is taken to be 20 kHz, although it may be defined

as 25kHz, 30kHz, 35kHz or even 40kHz. Each or any one of these thresholds may be taken as the human threshold of hearing when determining the threshold of human hearing in the context of the present invention.

Near audible noises may be defined in a similar manner, although in some instances near-audible noise may have a slightly lower frequency that may just be heard by some human listeners with particularly sensitive high-frequency hearing. Particularly preferred is near-audible noise that falls within a frequency range perceptible to a person with normal hearing, but at a sensitivity that makes its perception practically inaudible.

Preferably, the spoiling noise signal includes one or more components with centre frequencies in the range of from 17kHz to half the sampling frequency. Where there are multiple components, the spacing should preferably be in a range of from 1kHz to 5kHz. The centre frequency determines the frequency of the spoiling noise due to intermodulation effects, such as overtone alaising effects at frequencies equal to the difference between the frequency of the spoiling noise component and half the sampling rate. There is also the possibility of beat frequencies between the spoiling noise components. A particularly preferred embodiment has at least one component with a centre frequency in the range from 17 to 19kHz, at least one component in the range from 19 to 21kHz and at least one component in the range from 21kHz to half the sampling rate.

In addition to balanced maximum sampling rate encoded noise (e.g. 22050Hz alternate spikes), ultrasound and/or near audible sine wave noise bands of a predetermined level may be encoded to provide biasing or beating frequencies for specific recording devices or formats. For example, 17.2 kHz noise bands will provide a suitable bias/overtone for tape recorders using metal, chromium or 'normal' ferric tape, since the process of recording to tape is carried our using a bias frequency signal for a particular medium that is modulated by the base band signal to be recorded. As these frequencies are closer to the human hearing range, error-generated sub-harmonic noise (as before) may beat against these bias bands to create extra audible noise in frequency ranges that have lower audible perception levels and

are therefore more easily perceived as noise/sound. In effect, the error-produced noise is focussed into the frequency ranges most sensitive to the human ear. Many such bias bands may be found for varying devices and formats e.g. 20.7 kHz for PCM audio recording in camcorders and MiniDisc recorders/players, and 18.6 kHz for VHS Nicam recorders.

These central frequencies may be moved or rotated through as much as 400 Hz so as to modulate the beat produced, increasing the perceived harshness of the produced beat noise and increasing its complexity within the copy, making it more difficult to remove.

The invention is applied to a digital file but exploits effects in the analogue domain (whether actually carried out in the analogue domain or normal in digital processing).

The invention can be employed at any stage of the distribution of digital media - by physical media, by transfer of digital files, by streaming audio, by broadcast, live performances, or in public showings of recorded media. The noise signals can be added at the time of creation of the source signal, e.g. during editing, mastering or at the point of distribution.

The noise bias band and packet encoding may be generated by a program which may be given its encoding parameters in the form of a key. This key may be interpreted to provide the bias band frequencies, frequency rotation range, frequency rotation rate, bias noise start point within the output file, noise packet width, noise packet rampup/down rates, packet encoding start point and interval range, plus an interval sequence representing data or a function, as well as other appropriate information.

By means of such a key, a second program may reproduce the noise encoding so as to remove it when necessary. Alternatively or in addition, a digital audio file of the conjugate noise may be produced by either program for digital mixing and therefore cancellation of the encoded noise. Such keys or anti-noise files may be issued as part of a copyright control system.

In environments where it is never intended to remove such encoded noise, a

programmed sequence or random sequence realtime encoding program may be used to mix such a noise file at a concert, auditorium, cinema or such public address system so as to prevent quality copying of concert or cinema, etc. performances (for example by way of portable DAT recorders or video cameras) so as to help protect copyright material.

Re-sampling, re-recording and copying errors may increase as copies are made, increasing the noise audible within each subsequent copy. For example, noise levels may be low or inaudible at first copy to minidisk recorder, but may be noticeably audible on second copy. A particular advantage of this technique is that the increase of errors may be tailored so as to follow first copy, but not subsequent copies. In this way, a person may be allowed to make a personal copy without charge or at nominal charge, with the copyright owner knowing that the person will not be able to make high-quality subsequent copies for further, unauthorised distribution.

In systems where the digital audio file is delivered in a compressed form such as an MP3 or WMA file delivered over an Internet connection, the encoding scheme may anticipate the effect of the compression codec and compensate for this while encoding the inaudible noise, so that the correct encoding is achieved once the compression codec has been applied. In this way, the compressed file will be free of audible noise when replayed. The inaudible noise encoder may also be arranged so as to perform a similar encoding process on a compressed but otherwise unmodified version of the audio file.

On PC CD-ROM only or Multisession (CD-Audio or CD-ROM) audio CD, it may be possible for the CD-ROM data file to contain the audio file in an encrypted or key-enabled, self extracting file format. The inaudible noise-encoded file may either be delivered by the self-extraction program/process in its normal state as a file that may be interpreted and played as an audio file or as a stream (or series of data blocks comprising a stream) to an appropriate multimedia platform, e.g. Windows media player. Similarly the file or stream may be extracted by a program or process that may provide the audio file in its original form by cancelling or removing the inaudible noise by either generating the cancellation components from a provided key

or by digital mixing of an anti-noise file that corresponds to the particular audio file. In such systems, the noise-encoding key may be the same as the self-extraction encryption key, thereby allowing the user the benefit of having to provide only one such key to access his/her audio file.

In systems where the delivery method is known and fixed, it is also possible to allow the selective removal of inaudible noise components or to disable their encoding beforehand if so desired so as to optimise the level (and minimise quantization errors) for a subset of available noise components.

For audio files where the output level of the original audio is varied, the additional level contribution of the inaudible noise may be configured to represent the maximum available range of values when added, a calculated factor of the maximum available range or an additional value proportional to the original audio level. In this way it may be increasingly difficult for the noise to be removed by sampling and comparison of quiet areas of the audio. The level/ratio of inaudible noise added may also be so configured as to minimise any unwanted playback artefacts due to inadequacies of the playback platform so as to keep them at an acceptable or inaudible level.

Certain components of the inaudible noise may also be increased or decreased so as to minimise any unwanted artefact noise on one or more playback platforms.

Further, unwanted noise produced as artefacts due to limitations of accuracy on one or more preferred playback formats/platforms may be assessed or sampled so that the encoding process may compensate for or cancel such unwanted noise. This may maintain or improve the preferred playback quality, as compared to the uncompensated version, as required.

Other aspects of embodiments of the present invention are described and claimed below.

Brief Description of the Drawings

Embodiments of the present invention will now be described by way of

example only, with reference to the accompanying drawings in which:

Figure 1 shows a graph depicting the variation of human sensitivity to sound with both level and frequency;

Figure 2 illustrates schematically a first principle of embodiments of the present invention, that is, translation of energy from one frequency outside of the range of human hearing to a frequency within the range of human hearing;

Figure 3 shows a second principle used by embodiments of the present invention, that is, constructive interference or intermodulation products that combine to produce a signal within the range of human hearing that is audible;

Figure 4 shows a further principle upon which embodiments of the present invention can rely, that is, translation of energy at frequencies within human hearing (but imperceptible to the human ear) to a frequency within human hearing to form a perceivable frequency component;

Figure 5 shows, schematically, a process for protecting digital content according to an embodiment of the present invention;

Figure 6 shows a number of frequency spectrums for a number of such digitally encoded spoiling components;

Figure 7 shows a number of output spectra of the audio output by a Realistic minisette having recorded thereon the signal corresponding to the input spectra shown in Figure 6.

Figure 8 shows a further principle or aspect of embodiments of the present invention for creating spoiling frequency components within the range of human hearing.

Figure 9. shows an expanded portion of the noise packet described above with reference to Figure 8;

Figure 10 illustrates the effect produced by a noise packet due to an automatic level control response characteristic of an output device;

Figure 11 illustrates the effect of such packets in conjunction, for example, with the signals represented by the fifth spectrum shown in Figure 6;

Figure 12 depicts a number of output spectra, that is, post re-sampling output

spectra that correspond to 17.2kHz, 18.6kHz and 20.7kHz noise components shown in the first to third spectra of Figure 9;

Figure 13 illustrates a frequency spectrum of a signal output by a recording device following sampling of a base band signal that contains spoiling noise at 17.2kHz, 18.6kHz and 20.7kHz for a sampling rate of 22050 Hz;

Figure 14 depicts a further output spectrum produced from an audio signal output by an HP Pavilion N5461 laptop computer from a base band audio signal comprising 3 spoiling frequencies 17.2kHz, 18.6kHz and 20.7kHz with each of the centre frequencies having been rotated by 0.2kHz at a frequency of 2.2kHz; and

Figure 15 shows a third frequency spectrum 1800 for an audio signal output by the HP pavilion computer following re-sampling, at a frequency of 22050Hz, of an audio base band signal containing spoiling frequency components at 17.2kHz, 18.6kHz and 20.7kHz each of the frequencies of which were rotated by ±0.2kHz at a frequency of 2.2kHz together with the above described noise packets illustrated in and described with reference to figure 10.

Description of the Preferred Embodiments

Referring to Figure 1 there is shown a graph 100 that illustrates the human sensitivity to sound across a predetermined frequency range and for a range of loudness. The predetermined frequency range is, for the purposes of illustration, 0.02kHz to 20kHz. The loudness, measured in terms of decibels, is illustrated as varying from 0dB to 120dB. It can be appreciated that at very low and very high frequencies, sound of any magnitude is inaudible to the human ear. There is a region 102 between these very low and high frequencies where sound is completely inaudible below an inaudible sound level threshold 104 for respective volumes. There is a further region 106 where sound can be detected by the human ear but it is at or below the point or threshold 108 of perception. Sound that falls within a third region 110 is distinctly audible by the human ear. The audibility at approximately 2 to 2.5kHz represents a range of frequencies at which human hearing exhibits maximum sensitivity. Finally, there is a further region 112 at or in which pain may be

experienced by a listener. The audible region 110 and the further region 112 are separated by a boundary or threshold 114 that also varies with frequency.

Referring to figure 2 there is shown a first principle upon which embodiments of the present invention can be based. The arrangement 200 of figure 2 shows a frequency spectrum 202 of an input signal. The input signal is illustrated for the purpose of clarity only as comprising a single noise source 204 or spoiling frequency embedded within a base band signal (not shown). Also shown is a schematic representation of the frequency response 206 of a human ear. It can be appreciated that the cut-off point 208 of the frequency response of the human ear will be substantially 20kHz. A notional threshold 210 is also shown which corresponds to the threshold 108 as shown in Figure 1 at which sound becomes perceivable or audible. Again for the purpose of illustration only, this threshold 210 is shown as being substantially constant rather than varying with frequency as shown in Figure 1.

The signal having the frequency spectrum 202 shown in figure 2 is processed by a non-linear system 212 having a transfer function of H(w). The non-linear system 212 represents, for example, an analogue system that is used to re-record the sound output from an audio system (not shown), which sound has been derived from a digital source. As will be appreciated by those skilled in the art the non-linear system 212 will impose non-linear effects upon the signal having the frequency spectrum 202 illustrated. It can be seen that the frequency component 204 of the spoiling noise is beyond the frequency range of human hearing. Therefore, even though it is above the notional or schematic audible or perceivable threshold 210, it will be inaudible or imperceptible able and will not, therefore, detract from any listening pleasure. However, it can be appreciated from the right hand side of figure 2 that the frequency spectrum 214 of the output signal comprises a frequency component 216 that is within the range of human hearing and above the perceivable threshold level 210. It can be seen that the energy of the original frequency component 204 has been translated from that frequency component 204 to the frequency component 216 so that it is perceivable by human hearing. It will be appreciated that a number of non-linear

effects such as, for example, inter modulation products might give rise to such an energy transfer of energy from the first frequency component 204 to the second frequency component 216.

Figure 3 shows an arrangement 300 similar to that shown in figure 2 for creating spoiling noise 302 in the spectrum 304 of an output signal from a pair 306 and 308 of frequency components of the input spectrum 310 of an input signal (not shown). It can be seen from the frequency spectrum 310 of the input signal that the pair 306 and 308 of frequency components both lie below the perceivable threshold 210. Therefore, even though these frequency components are present in the audible output and are both within the audible frequency range, they do not interfere with the listeners listening pleasure.

However, when the audible output is processed by a non-linear system 312, it can be appreciated that the energy formerly associated with the pair of frequency components 306 and 308 is transferred, by inter- modulation and constructive interference, it is, by non-linear effects, to a further frequency component 314 that is located both within the frequency spectrum of human hearing 206 and such that it has a magnitude that is greater than the audible or perceivable threshold 210. Therefore, when the signal output from, for example, a Hi-fi system that is playing a CD which contains the conventional base band signal data from which conventional base bands signals for music can be derived and the spoiling components 306 and 308, the non-linear system 312 used to re-record the audible output creates a spoiling frequency component 314 that cannot be filtered without adversely effecting the reproduced base band signal. It can be appreciated that re-recording audio produced in accordance with embodiments of the present invention can be used to spoil or detract from a listener's listening pleasure.

Referring to figure 4, there is shown an arrangement 400 for producing a spoiling frequency 402 from a pair 404 and 406 of spoiling frequencies contained

within the output of a Hi-fi system (not shown) in response to playing digital media. It can be appreciated that the spoiling frequencies 404 and 406 are both within the range of human hearing but they are also both below the perceivable threshold 210 and, as such, do not interfere with the listening pleasure of a listener.

If a non-linear system such as, for example, a tape recorder 408, is used to re-record the audio, the spoiling frequencies 404 and 406 have been selected to exploit the non-linearities of the non-linear system 408 to ensure that a relatively sizeable spoiling frequency 402 is produced, via, for example, inter-modulation products, due to the transfer of energy from the spoiling frequency components 404 and 406 to that further spoiling frequency 402. It can be appreciated that the spoiling frequency 402 has a magnitude that is greater than the perceivable human hearing perceivable threshold 210 and that it falls within the frequency spectrum 206 of human hearing.

Referring to figure 5, there is shown, schematically a process 800 for protecting digital content according to an embodiment of the present invention. A base band audio signal 802 is combined with a source of spoiling noise (not shown). The spoiling noise has a frequency spectrum as shown by 804. The audio base band signal and the spoiling noise are combined to produce digital content which, in the illustrated example, is shown as having been stored on a CD 806. When the digitised data is processed by a Hi-fi system 808, it produces, via the loud speakers 810, audio having an output spectrum 812 that comprises a faithful reproduction 814 of the original audio 802 together with a spoiling frequency component 816 that is derived from the spoiling noise shown in the spoiling noise spectrum 804. It can be appreciated that the spoiling noise or frequency component 816 falls outside of the frequency response of the human ear 818.

However, if the audio output signal is reprocessed using, for example, a tape recorder or captured and reprocessed using a PC soundcard 820, the spoiling noise

frequency component 816 has at least one of its frequency and magnitude selected to create, within the frequency response of the human ear 818, an undesirable artefact or frequency component 822.

An example of the application of the above principles of embodiments of the present invention will now be described with reference to protecting digital content against illegitimate copying using a tape recorder. In the present example the tape recorder was a "Realistic minisette - 20" tape recorder which used an ordinary ferric oxide tape and the recording relied upon the built in electret condenser microphone. A blank digital audio sound file was created as the encoded source or digital content. The blank digital audio sound file only contained the spoiling components or sources of spoiling frequency components to allow the effect of each component to be assessed in the output spectrum of any signals derived from the illegitimate copy created using the realistic minisette.

Figure 6 shows a number of frequency spectrums 900 for a number of such digitally encoded spoiling components. The first spectrum 902 comprises a frequency component 904 having a centre frequency of 17.2kHz and a magnitude of about -10 dB to -12 dB. In preferred embodiments, the magnitudes are at least -20 dB and greater. A second spectrum 906 is shown for a signal having a frequency component 908 that is centred on 18.6kHz. The magnitude of this frequency component 908 is about -10 dB to -12 dB. In preferred embodiments, the magnitude is at least -20 dB and greater. A third frequency spectrum 910 is shown as having a frequency component 912 centred on 20.7kHz and having a magnitude of -10 dB to -12 dB. In preferred embodiments, the magnitude is at least -20 dB and, preferably, greater.

A fourth frequency spectrum 914 is also illustrated. The fourth frequency spectrum 914 comprises 3 frequency components 916 to 920 that are centred on 17.2kHz, 18.6kHz and 20.7kHz. In effect, the fourth frequency spectrum 914 is a combination of the first three spectra 902, 906 and 910. Also shown is sixth frequency spectrum 922 that represents a signal having 3 frequency components 924

to 928 on 17.2kHz, 18.6kHz and 20.7kHz respectively. Preferably, the centre frequency of each frequency component is arranged to oscillate about the centre frequencies of 17.2kHz, 18.6kHz and 20.7kHz by between ±0.1 kHz and 0.4 kHz. In preferred embodiments, the oscillation is performed at a frequency of about 1kHz to 5kHz e.g. 2.2 kHz.

Each of the signals corresponding to the frequency spectrums 902, 906, 910, 914 and 922, having been output via a Woolworth's CD, model T-295, were recorded using the Realistic minisette 20 tape recorder to identify the effect of the frequency components 904, 908, 912, 916, 918, 920, 926 and 928.

Figure 7 shows a number of output spectra 1000 of the audio output by the realistic minisette having recorded the signals corresponding to the input spectra shown in Figure 6. It will be appreciated that the spectra shown in Figure 6 are input spectra from the perspective of the device used to re-record or re-process the signals corresponding to those spectra. However, the spectra shown in Figure 6 are output spectra when viewed from the perspective of the legitimate reproduction device. The first output spectra 1002 is the spectrum produced having recorded the first input spectrum 902 with its frequency component of 17.2kHz. It can be appreciated that much of the energy of the 17.2kHz frequency component 904 has been redistributed to relatively low frequency and low magnitude frequency components. The magnitude of the spoiling components are between -45 dB and -55 dB for frequencies of approximately 50 Hz to 3.1 kHz together with a -45 dB peak at 4.6 kHz.

The second output spectrum 1006 corresponds to the audio signal produced in response to the tape recorder having recorded the second input spectrum 906 with its 18.6kHz frequency component 908. It can be appreciated that the spectrum 1006 comprises a frequency components having magnitudes of -45 dB to -55 dB for frequencies of 60 Hz to 2 kHz together with a relatively strong -32 dB component at approximately 3.5 kHz.

It can be appreciated that a third output spectrum 1010, which corresponds to the third input spectrum 910, has a number of frequency components 1012 that are distributed over frequency range of 40 Hz to 3.5 kHz with magnitudes of between -45 dB and -55 dB together with a relatively strong -36 dB peak at approximately 1.5 kHz. It will be appreciated that this spoiling noise or these spoiling frequency components 1012 fall squarely within the range of maximum human hearing sensitivity and have a relatively large magnitude, especially as compared to the components generated by the 17.2kHz signal.

A fourth output spectrum 1014 is shown. The fourth output spectrum 1014 is derived from the signal having the fourth spectrum 914 as shown in figure 6. It can be seen that the three frequency components 916 to 920 of that fourth spectrum 914 have produced a significant number of intermodulation products that are distributed over a frequency range of 5.4 kHz to 14.3 kHz. A number of the more significant inter modulation products 1016 are distributed over a frequency range of about 1 Hz to 5.4kHz.. It can be appreciated that the main frequency components 1018 to 1026 will represent significant spoiling noise that will adversely effect the listening pleasure of any illegitimate recording of an audio base band signal having spoiling frequency components 916 to 920.

A fifth frequency spectrum 1028 corresponding to the fifth input spectrum 922 is also shown in figure 7. Again, it can be appreciated that the fifth spectrum 1028 has a significant number of inter-modulation products 1030 that fall within the frequency response of the human hearing. However, as compared to the fourth spectrum 1014, it can be appreciated that the higher frequency inter-modulation products 1032 have a much smaller magnitude as compared to corresponding inter-modulation products 1034 of the fourth output spectrum 1014. It can be seen that much of the energy associated with the high frequency inter-modulation products 1032 has been re-focussed to lower frequencies, which has resulted in much greater average magnitudes of the frequency components in these lower frequencies.

Referring to figure 8, there is shown a further principle or aspect of embodiments of the present invention for creating spoiling frequency components within the range of human hearing. Figure 8 shows a blank section of a digital base band signal 1102 that has been provided with noise in accordance with embodiments of the present invention. In particular, the main body 1104 of the signal comprises sine wave noise band encoding which, as described above, takes the form of 3 sine wave components having respective frequencies of 17.2kHz, 18.6kHz and 20.7kHz together with an additional 22.05kHz noise component. At the end of the sine wave noise band encoding section 1104, the sine wave component level is ramped down or reduced to zero as indicated by the decreasing portions 1106 of the signal 1102 prior to the introduction of a noise packet 1108 that has a very specific shape and, in preferred embodiments, comprises solely 22.05kHz noise. Preferably, the noise packets 1108 are regularly distributed in time throughout the base band signal 1102. After each noise packet 1108, the sine wave component level is ramped up or increased as indicated by signal portions 1110.

Figure 9 shows an expanded portion 1202 of the noise packet 1108 described above with reference to figure 8.

It will be appreciated that figures 8 and 9 show the base band signal 1102 as not containing any recorded audio information. This is in the interest of clarity and to show clearly how the noise signals are encoded. Recorded audio information is simply superimposed on and/or mixed into the noise signal shown in figures 8 and 9.

Figure 10 illustrates the effect produced by a noise packet due to an automatic level control response characteristic of, for example, the Realistic minisette 20 tape recorder. The noise band or noise packet 1108 has a substantially symmetrical geometrical form or envelope, ramping up from 0 to a plateau 1302 and then ramping back down to 0. The solid lines 1304 show the ALC level output (a modulation) of a typical ALC circuit, giving a fast attack response to a peak leading to a fast level

reduction 1306, followed by a slow decay after the peak, which then leads to a gradual level increase 1308. It will be seen that, in contrast to the symmetrical geometry of the noise packet 1108 in the base band signal 1102, the ALC response is geometrically asymmetrical. The symmetrical noise packets 1108 have sufficiently slow rates of change such that they do not produce any significant perceivable audio sound when reproduced from the original digital content. However, when the symmetry is broken due to modification by the ALC modulation, the modulated sound packet output introduces a significant amount of noise at an audio frequency. This modulation has been found to approximate to a continuous audio output level reduction if the noise packets 1108 are sufficiently frequently distributed throughout the base band signal 1102. It will be appreciated by those skilled in the art that such a significant reduction in the average audio output level due to the packets 1108 will interfere with any listening pleasure associated with playing an illegitimate copy of digital content having such a signal 1102 embedded therein or associated therewith.

Figure 11 shows the effect of such packets 1108 in conjunction, for example, with the signals represented by the fifth spectrum 922 shown in figure 6. It can be seen that the response, is very similar to the output spectrum 1028 with the exception of a slightly raised magnitude at about 40Hz. Figure 11 shows a number of spectra 1400 for an embodiment that uses both the noise packets 1108 together with the signal having the rotating or oscillating frequency components 924 to 928 shown in the fifth spectrum 922. The output spectrum produced by such an embodiment is shown the third spectrum 1402 of the figure 11. The other two spectra correspond to the fourth and fifth spectra 1014 and 1028 respectively of figure 7 and are included for comparison purposes. It can be appreciated from the third output spectrum 1402 that there is a region 1404 having a significantly increased average power level.

A further application of embodiments of the present invention will now be described with reference to the protection of digital content stored on a CD-audio disc. In the example, the digital content stored on the CD was played on a

Woolworth's T-9295 personal CD player, which is a typical personal CD player. The reproduced audio signal was re-sampled using an HP Pavilion N5461 laptop computer via the audio input connection of its built in sound card at a CD quality sampling rate of 44.1kHz using 16 bit stereo resolution. The sampled audio was replayed and its spectrum was analysed.

Figure 12 shows a number of output spectra 1500, that is post re-sampling output spectra 1502, 1504 and 1506, that correspond to the 17.2kHz, 18.6kHz and 20.7kHz noise components shown in the first to third 902, 906 and 910 spectra of figure 6.

It can be appreciated that each frequency component 904, 908 and 912 clearly manifests itself as one or more sub-harmonic tones. The magnitude of the sub-harmonic tones progressively increases with frequency but shows a marked roll off in amplitude at 20.7kHz. When each of the frequency components is used singularly, the tones produced are wholly unacceptable and manifest themselves as noise that is clearly heard as sub-tones.

Referring to figure 13 there is shown a frequency spectrum of a signal output by a recording device following sampling of a base band signal that contains spoiling noise at 17.2kHz, 18.6kHz and 20.7kHz for a sampling rate of 22050 Hz. The 17.2kHz, 18.6kHz and 20.7kHz signal components can be clearly seen at reference numerals 1602, 1604 and 1606 respectively. It can be seen that due to at least one of inter modulation between these frequency components 1602 to 1606 and due to aliasing effects a significant number of harmonics 1608, for example, are produced well within the audio range of human hearing.

Referring to figure 14 there is shown a further output spectrum 1700 produced from an audio signal output by the HP Pavilion N5461 laptop computer from a base band audio signal comprising 3 spoiling frequencies 1702, 1704 and 1706 that

correspond to frequencies 17.2kHz, 18.6kHz and 20.7kHz respectively with each of the centre frequencies having been rotated by 0.2kHz at a frequency of 2.2kHz. It can be appreciated that, again, a significant number of inter modulation products and a significant degree of aliasing has resulted which causes undesirable frequency components 1708 to fall within the range of human hearing. It can be appreciated that the undesirable components 1708 have slightly greater magnitude when compared to the corresponding components 1608 of figure 13.

Referring to figure 15 there is shown a third frequency spectrum 1800 for an audio signal output by the HP pavilion computer following re-sampling, at a frequency of 22050Hz, of an audio base band signal containing spoiling frequency components at 17.2kHz, 18.6kHz and 20.7kHz each or this centre frequencies of which were rotated by ± 0.2kHz at a frequency of 2.2kHz together with the above described noise packets illustrated in and described with reference to figure 10. It can be appreciated from the output spectrum 1800 that the 17.2kHz, 18.6kHz and 20.7kHz components 1802, 1804 and 1806 respectively have relatively large magnitudes. It can also be appreciated that there is a region 1808 of appreciable spoiling noise derived from at least one of aliasing of the 17.2kHz, 18.6kHz and 20.7kHz signals and inter modulation products derived from those frequencies, as well as, further aliasing caused by in adequate sampling of those inter modulation products. It can also be appreciated that there is a further region 1810 of appreciable noise that is between approximately 12kHz and 16kHz. The noise illustrated in the output spectrum 1800 is significant and will detract from the listening pleasure of any listener.

It will be appreciated by those skilled in the art that the spoiling noise created and described with reference to the above embodiments results from at least one of the following processes

1. intermodulation products being generated from the spoiling frequency components 17.2, 18.6 and 20.7kHz signals;

2. aliasing due to an inadequate sampling frequency, that is, due to sampling at frequencies below the Nyquist frequency;

3. Moiré noise generated by, again, inadequate sampling frequencies;

4. frequencies rotation of the spoiling components at at least one of 17.2kHz, 18.6kHz and 20.7kHz as well associated inter modulation resulting from such rotation;

5. high and low frequency noise components resulting from constructive and destructive interference between components generated via the above processors; and

6. packet asymmetry noise.

In preferred embodiments, the frequency rotation, that is, the oscillations of the centre frequency of one or more of the spoiling noise components at 17.2kHz, 18.6kHz and 20.7kHz is oscillated by ± 0.1kHz and ± 0.4kHz and is preferably oscillated by 0.2kHz. The frequency of oscillation is preferably between 0.01 to 0.05 x the sampling frequency which, for the CD format, where the sampling frequency is 22050Hz, gives a frequency range of 0.22kHz to 11kHz. In the embodiments described above, the oscillation was 2.2kHz, that is, the frequency used was the sampling frequency (22050Hz) x 0.1.

Although the above embodiments have described with reference to producing a CD, embodiments are not therefore limited thereto. Embodiments can equally well be realised in which DVD, digital magnetic media and computer files are created using the above.

Furthermore, although the embodiments described above have used substantially symmetrical noise packet, embodiments are not limited to such an arrangement. Embodiments can be realised that use substantially asymmetrical noise packets.

Whilst we have described exemplary embodiments of the present invention it will be appreciated that the preceding description is intended to be illustrative rather than restrictive and the invention is not to be limited except by the scope of the

appended claims.

Variations of the described embodiments may be made. For example the parameters of the spoiling noise components may be calculated on the basis of other sampling frequencies than the 44.1kHz for CD for example 32kHz, 48 kHz, 48.3 kHz, 88.2 kHz, 96kHz, 192kHz.

CLAIMS

1. A method of creating a digital signal representing an audio signal that is resistant to copying, the method comprising:

receiving a source digital signal having a predetermined sampling frequency;

generating a spoiling noise signal having a centre frequency below half said predetermined sampling frequency, said spoiling noise signal being imperceptible to a human with normal hearing if reproduced faithfully and having a frequency and amplitude such that when transformed with a non-linear transformation a human-perceptible noise is created; and

generating an output digital signal that is resistant to copying by combining said source digital signal and said spoiling noise signal.

2. A method according to claim 1 wherein said centre frequency of said spoiling noise signal oscillates within a range of from 100 to 400Hz.

3. A method according to claim 2 wherein said centre frequency of said spoiling noise signal oscillates with a frequency in the range of from 1000Hz to 5000Hz.

4. A method according to claim 2 wherein said centre frequency of said spoiling

noise signal oscillates with a frequency in the range of from 1/10 to 1/2 of said predetermined sampling frequency.

5. A method according to any one of the preceding claims wherein said spoiling noise signal comprises first and second spoiling components such that inter-modulation components of said first and second components create said human-perceptible noise.

6. A method according to any one of the preceding claims wherein said spoiling noise signal comprises a carrier signal of above or near-audible frequency modulated by a modulation envelope, said modulation envelope having the form of a repeated substantially symmetric packet.

7. A method according to claim 6 wherein said packet is substantially trapezoidal in shape.

8. A method according to claim 6 or 7 wherein the rate of change of said modulation envelope is sufficiently slow as to not cause any audible sound when said output signal is faithfully reproduced.

9. A method according to claim 8 wherein the ratio of the rate of change of the rising part of said modulation envelope to the width of said packet is in the range of from 1/3 to 1/10, preferably in the range of from 1/3 to 1/5.

10. A method according to any one of claims 6 to 9 wherein said packet has a width in the range of from 100 to 1000 samples.

11. A method according to any one of claims 6 to 10 wherein the inverse of the duration of said packet is a human audible frequency.

12. A method according to claim 11 wherein the inverse of the duration of said packet is in the range of from 1 to 5kHz.

13. A method according to any one of claims 6 to 12 wherein said modulation envelope defines multiple packets of different lengths, geometries and/or intervals.

14. A method according to any one of the preceding claims wherein said spoiling noise signal has one or more components having centre frequencies in the range of from 17kHz to 22.050kHz, preferably within 400Hz of at least one of 17.2kHz, 18.6kHz and 20.7kHz.

15. A method according to any one of the preceding claims wherein said spoiling noise signal comprises a plurality of noise components and wherein the spacing between the centre frequencies of different ones of said noise components is in the range of from 1kHz to 5kHz.

16. A method according to any one of the preceding claims wherein said spoiling noise signal has a component that is calculated for a given target device to cause said human-perceptible noise due to a known non-linearity of said target device.

17. A method according to claim 16 wherein said given target device is a generic device representing characteristics of a class of devices.

18. A method according to claim 16 or 17 wherein said known non-linearity is at least one characteristic selected from the group comprising:

frequency dispersion,
frequency non-linearities,
phase dispersion,
timing aberration,
quantisation errors,

lossy characteristics of a codec,
psychoacoustic effects of a codec,
analog clipping, and
aliasing in resampling.

19. A method according to claim 16 or 17 wherein said component of said spoiling noise signal has a frequency that is calculated to cause said human-perceptible noise by forming beat frequencies with a noise signal created by said target device.

20. A method according to any one of claims 1 to 19 further comprising:
recording said output digital signal on a recording medium.

21. A method according to claim 20 wherein said recording medium is selected from the group comprising:
CD-Audio, CD-ROM, CD-RAM, MiniDisc, Laser Disc, DVD, DAT, DCC.

22. A method according to any one of claims 1 to 19 further comprising:
forming said output digital signal into a computer-readable file.

23. A method according to claim 22 wherein said computer-readable file conforms to a format selected from the group comprising:
mp3, ATRAC, Windows Media, .avi, .wav,, MIDI, AAC, Macintosh AIFF resource.

24. A method according to any one of claims 1 to 19 further comprising:
transmitting said output digital signal over a transmission channel.

25. A method according to claim 25 wherein said transmission channel is selected from the group comprising:

terrestrial radio broadcast, terrestrial television broadcast, DAB, satellite radio, satellite television, cable television, the Intranet, an intranet, an extranet, streaming audio, a telephone network.

26. A method according to any one of claims 1 to 19 further comprising:
reproducing said output digital signal as an audio signal.

27. A method according to any one of the preceding claims wherein said source digital signal is a signal selected from the group comprising:
a musical composition, a motion picture soundtrack, a game soundtrack, a television programme soundtrack, a live performance.

28. A computer program comprising program code means that, when executed on a computer system, instruct the computer system to effect the method of any one of the preceding claims.

29. A computer-readable storage medium having stored thereon a computer program according to claim 28.

30. A digital signal representing an audio signal that is resistant to copying, the signal comprising:
a source digital signal representing a desired audio signal and having a predetermined sampling frequency;
spoiling noise signal having a centre frequency below half said predetermined sampling frequency, said spoiling noise signal being imperceptible to a human with normal hearing if reproduced faithfully and having a frequency and amplitude such that when transformed with a non-linear transformation a human-perceptible noise is created.

31. A digital signal according to claim 30 wherein said centre frequency of said

spoiling noise signal oscillates within a range of from 100 to 400Hz.

32. A digital signal according to claim 31 wherein said centre frequency of said spoiling noise signal oscillates with a frequency in the range of from 1000Hz to 5000Hz.

33. A digital signal according to claim 31 wherein said centre frequency of said spoiling noise signal oscillates with a frequency in the range of from 1/10 to 1/2 of said predetermined sampling frequency.

34. A digital signal according to any one of claims 30 to 33 wherein said spoiling noise signal comprises first and second spoiling components such that inter-modulation components of said first and second components create said human-perceptible noise.

35. A digital signal according to any one of claims 30 to 34 wherein said spoiling noise signal comprises a carrier signal of above or near-audible frequency modulated by a modulation envelope, said modulation envelope having the form of a repeated substantially symmetric packet.

36. A digital signal according to claim 35 wherein said packet is substantially trapezoidal in shape.

37. A digital signal according to claim 35 or 36 wherein the rate of change of said modulation envelope is sufficiently slow as to not cause any audible sound when said output signal is faithfully reproduced.

38. A digital signal according to claim 37 wherein the ratio of the rate of change of the rising part of said modulation envelope to the width of said packet is in the range of from 1/3 to 1/10, preferably in the range of from 1/3 to 1/5.

39. A digital signal according to any one of claims 35 to 38 wherein said packet has a width in the range of from 100 to 1000 samples.

40. A digital signal according to any one of claims 35 to 39 wherein the inverse of the duration of said packet is a human audible frequency.

41. A digital signal according to claim 40 wherein the inverse of the duration of said packet is in the range of from 1 to 5kHz.

42. A digital signal according to any one of claims 35 to 41 wherein said modulation envelope defines multiple packets of different lengths, geometries and/or intervals.

43. A digital signal according to any one of the preceding claims wherein said spoiling noise signal has one or more components having centre frequencies in the range of from 17kHz to 22.050kHz, preferably within 400Hz of at least one of 17.2kHz, 18.6kHz and 20.7kHz.

44. A digital signal according to any one of the preceding claims wherein said spoiling noise signal comprises a plurality of noise components and wherein the spacing between the centre frequencies of different ones of said noise components is in the range of from 1kHz to 5kHz.

45. A digital signal according to any one of claims 30 to 44 wherein said source digital signal is a signal selected from the group comprising:

a musical composition, a motion picture soundtrack, a game soundtrack, a television programme soundtrack.

46. A digital signal according to any one of claims 30 to 45 wherein said digital

signal is in the form of a computer-readable file.

47. A digital signal according to claim 37 wherein said computer-readable file conforms to a format selected from the group comprising:

mp3, ATRAC, Windows Media, .avi, .wav,, MIDI, AAC, Macintosh AIFF resource.

48. A recording medium having recorded thereon a digital signal according to any one of claims 30 to 47.

49. A recording medium according to claim 48 wherein said recording medium is selected from the group comprising:

CD-Audio, CD-ROM, CD-RAM, MiniDisc, Laser Disc, DVD, DAT, DCC.

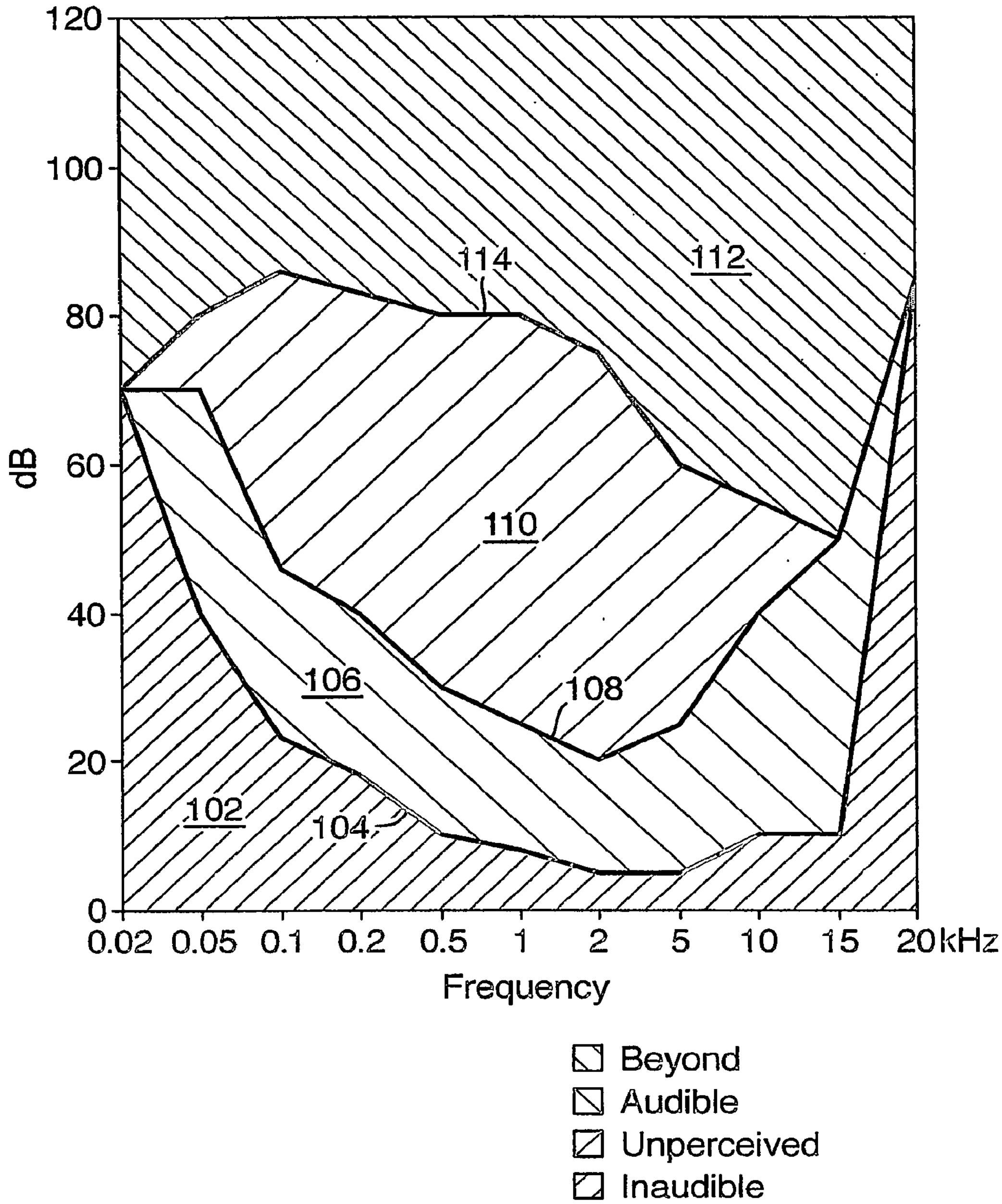
50. An apparatus for creating a digital signal representing an audio signal that is resistant to copying, the apparatus comprising:

an input circuit to receive a source digital signal having a predetermined sampling frequency;

a signal generator to generate a spoiling noise signal having a centre frequency below half said predetermined sampling frequency, said spoiling noise signal being imperceptible to a human with normal hearing if reproduced faithfully and having a frequency and amplitude such that when transformed with a non-linear transformation a human-perceptible noise is created; and

an adder for generating an output digital signal that is resistant to copying by adding said source digital signal and said spoiling noise signal.

Fig.1.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80
60
40
20
0
dB
0.02
0.05
0.1
0.2
0.5
1
2
5
10
15
20kHz
Frequency
114
112
110
106
108
102
104
Beyond
Audible
Unperceived
Inaudible

Fig.2.

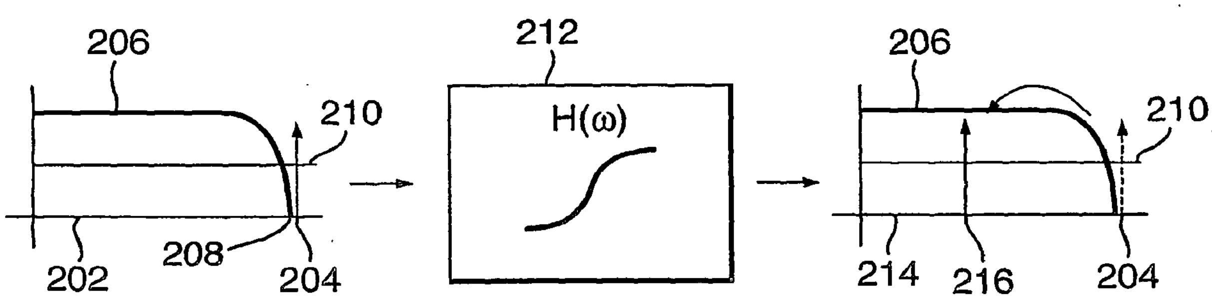


Fig.3.

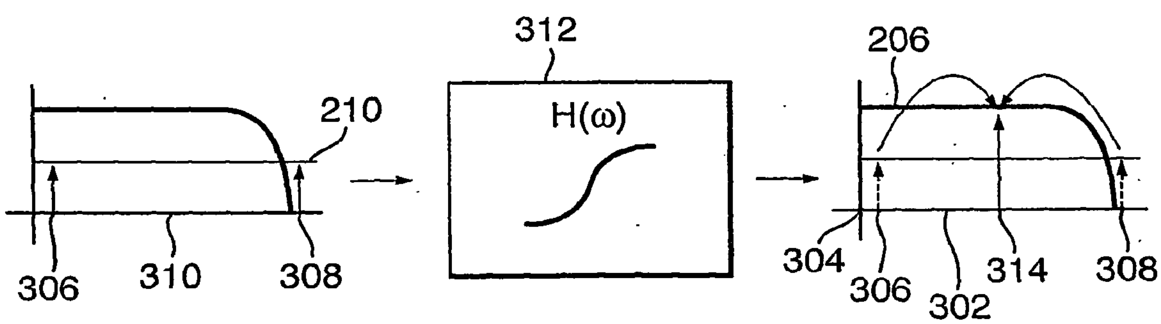


Fig.4.

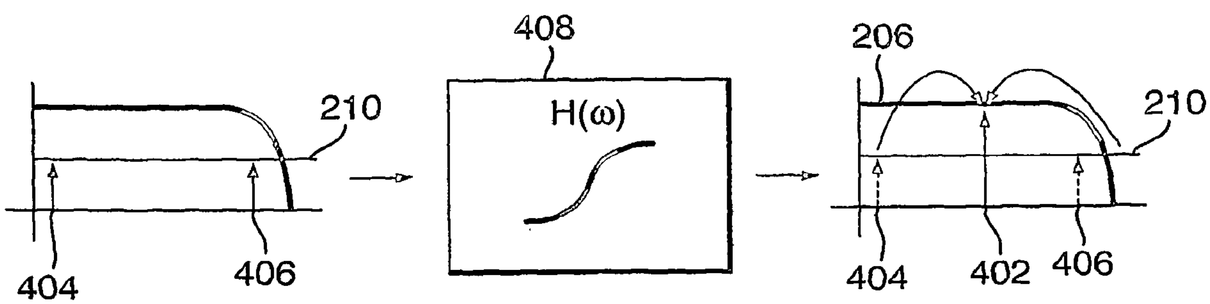


Fig.5.

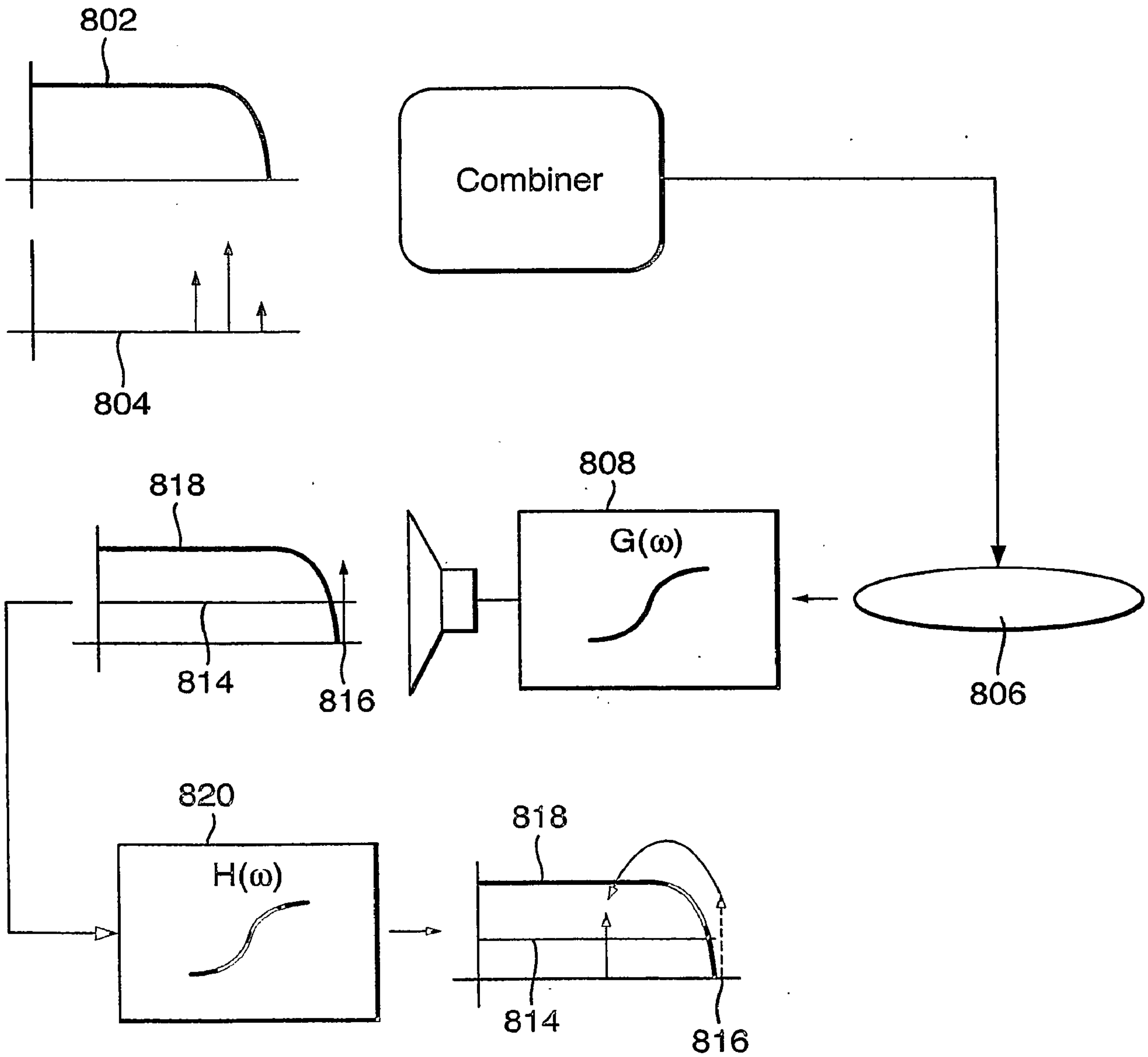


Fig.6.

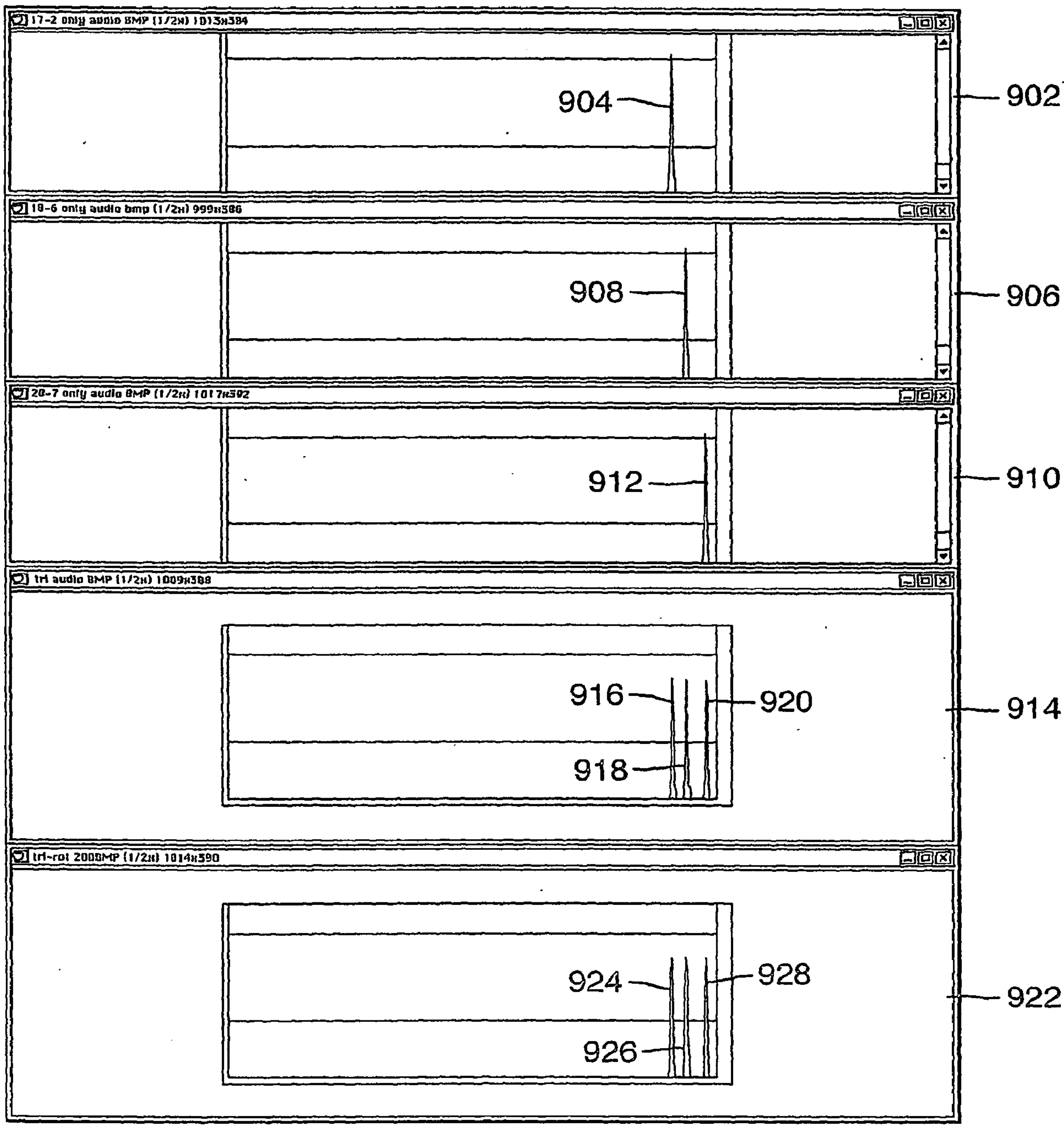


Fig.7.

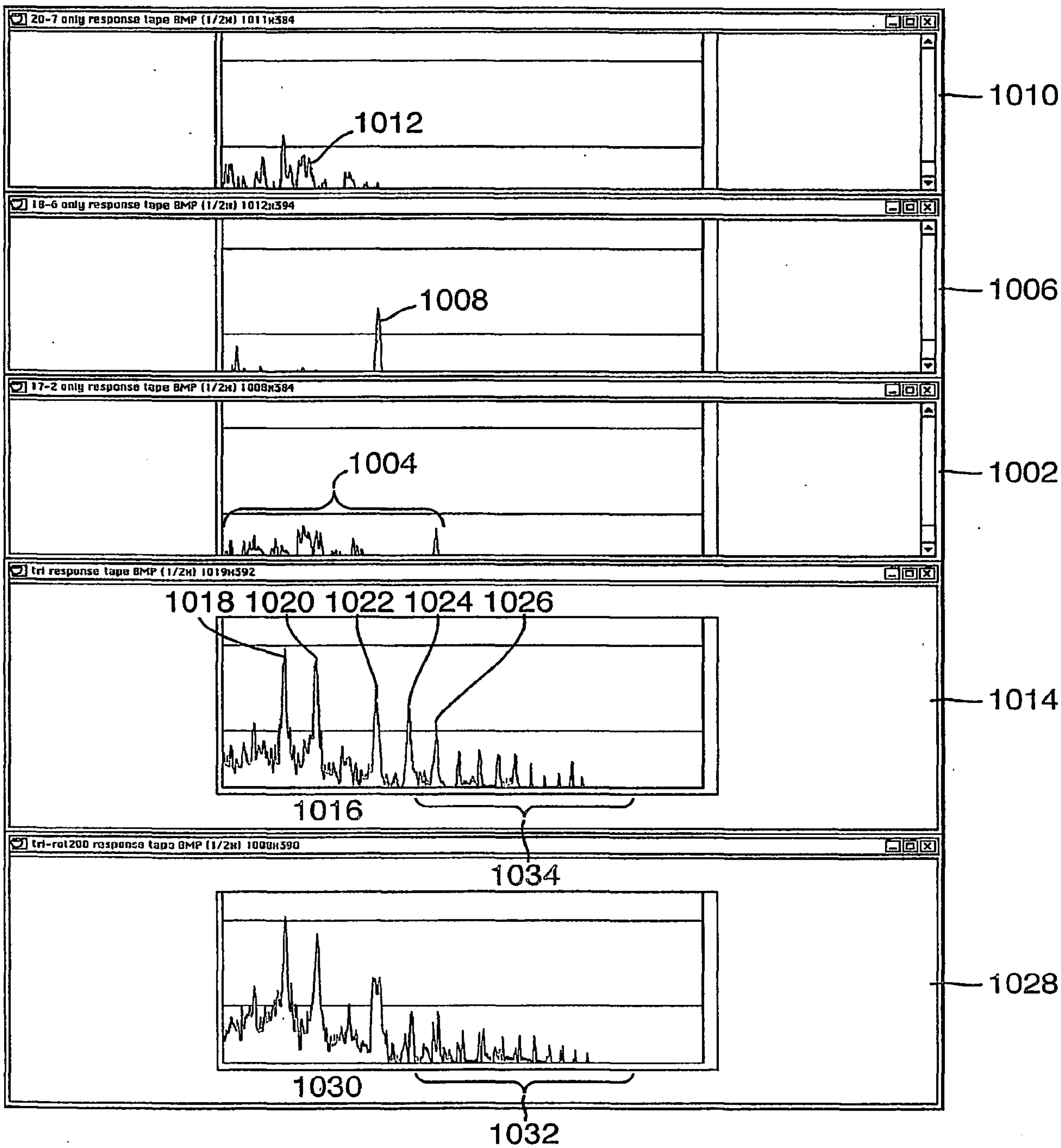

1010
1012
1006
1008
1002
1004
1018 1020 1022 1024 1026
1014
1016
1034
1028
1030
1032

Fig.8.

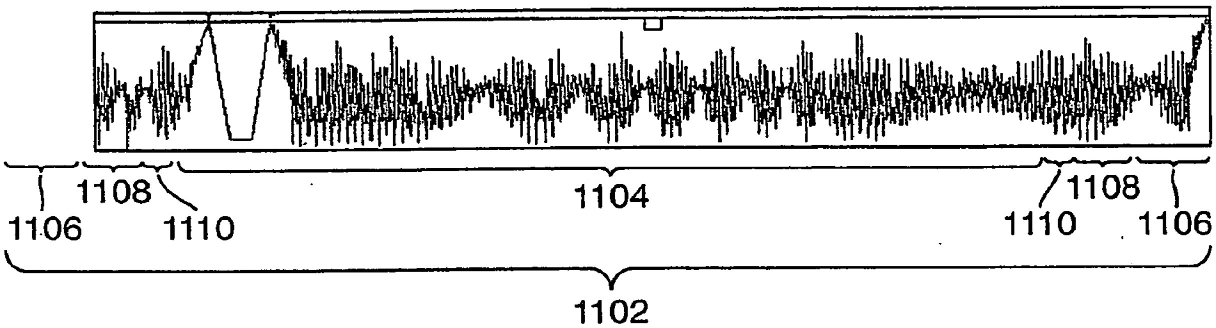


Fig.9.

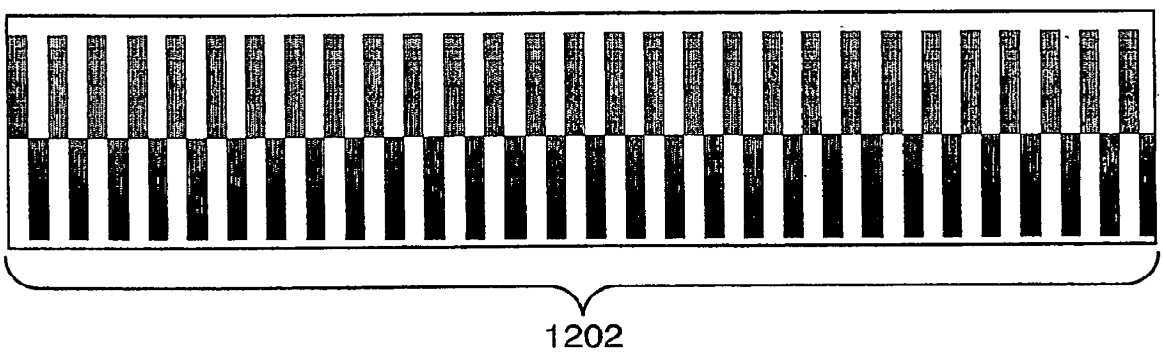


Fig.10.

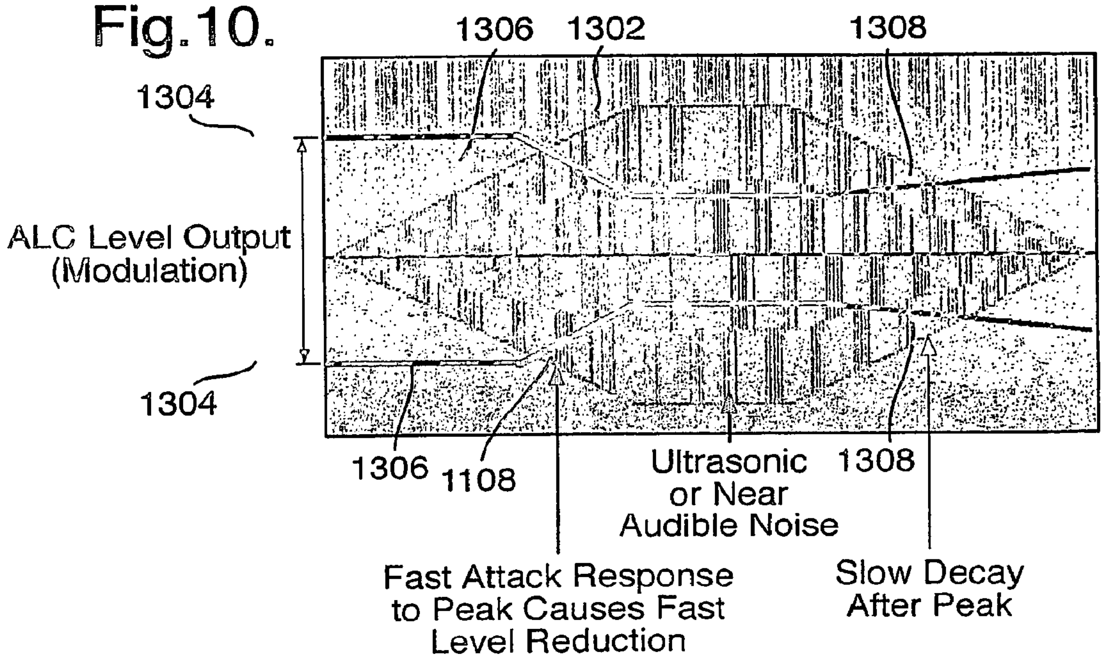


Fig.11.

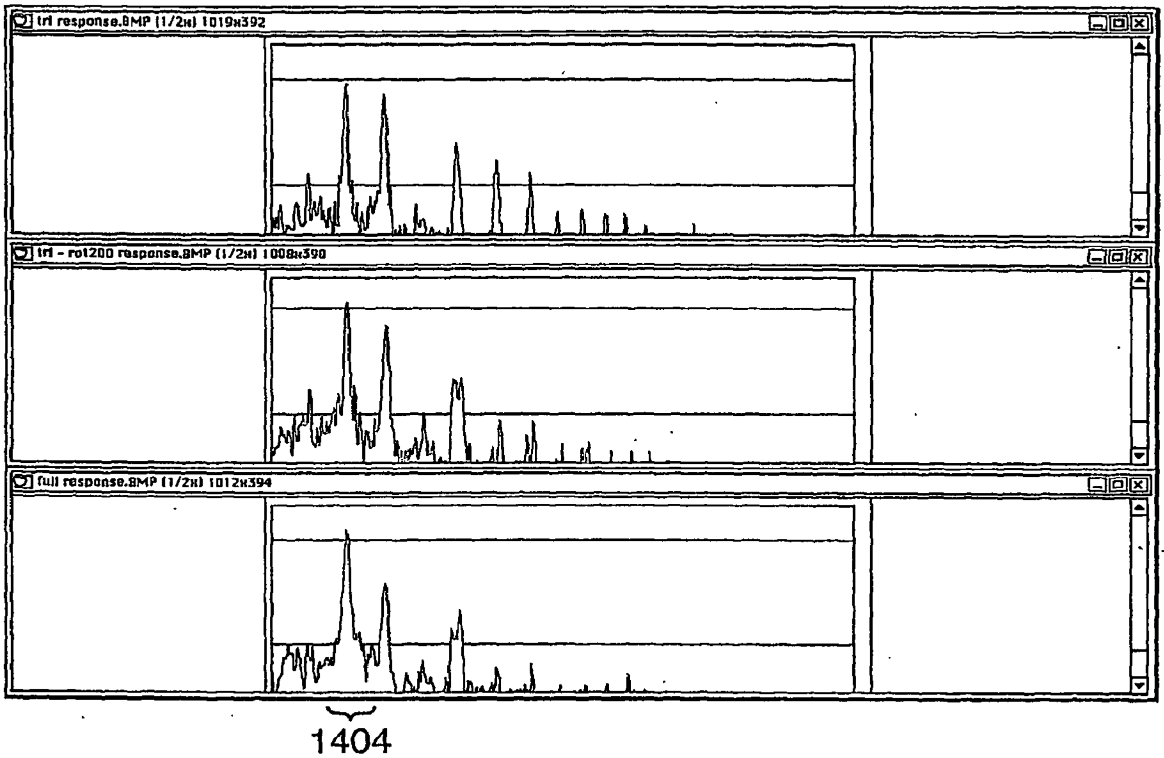
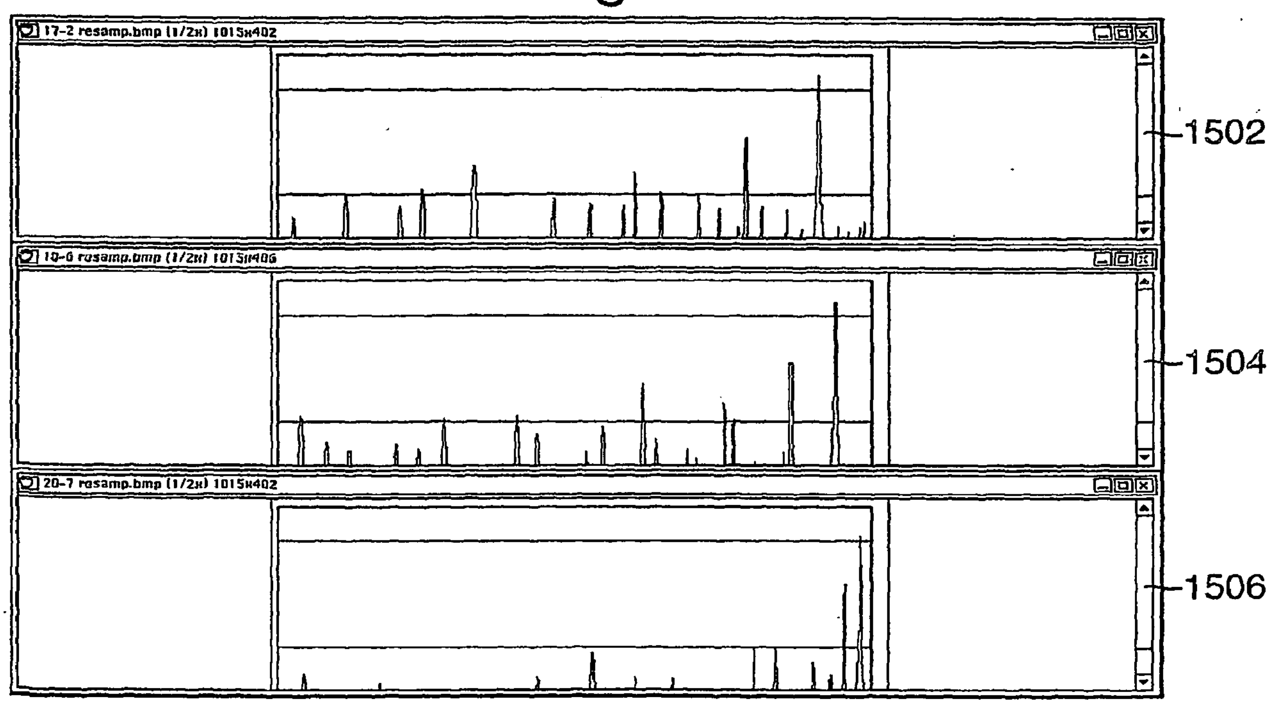

trl response.BMP (1/2x) 1019x392
trl - rot200 response.BMP (1/2x) 1008x390
full response.BMP (1/2x) 1012x394
1404

Fig.12.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1502
10-6 resamp.bmp (1/2x) 1015x406
1504
20-7 resamp.bmp (1/2x) 1015x402
1506

Fig.13.

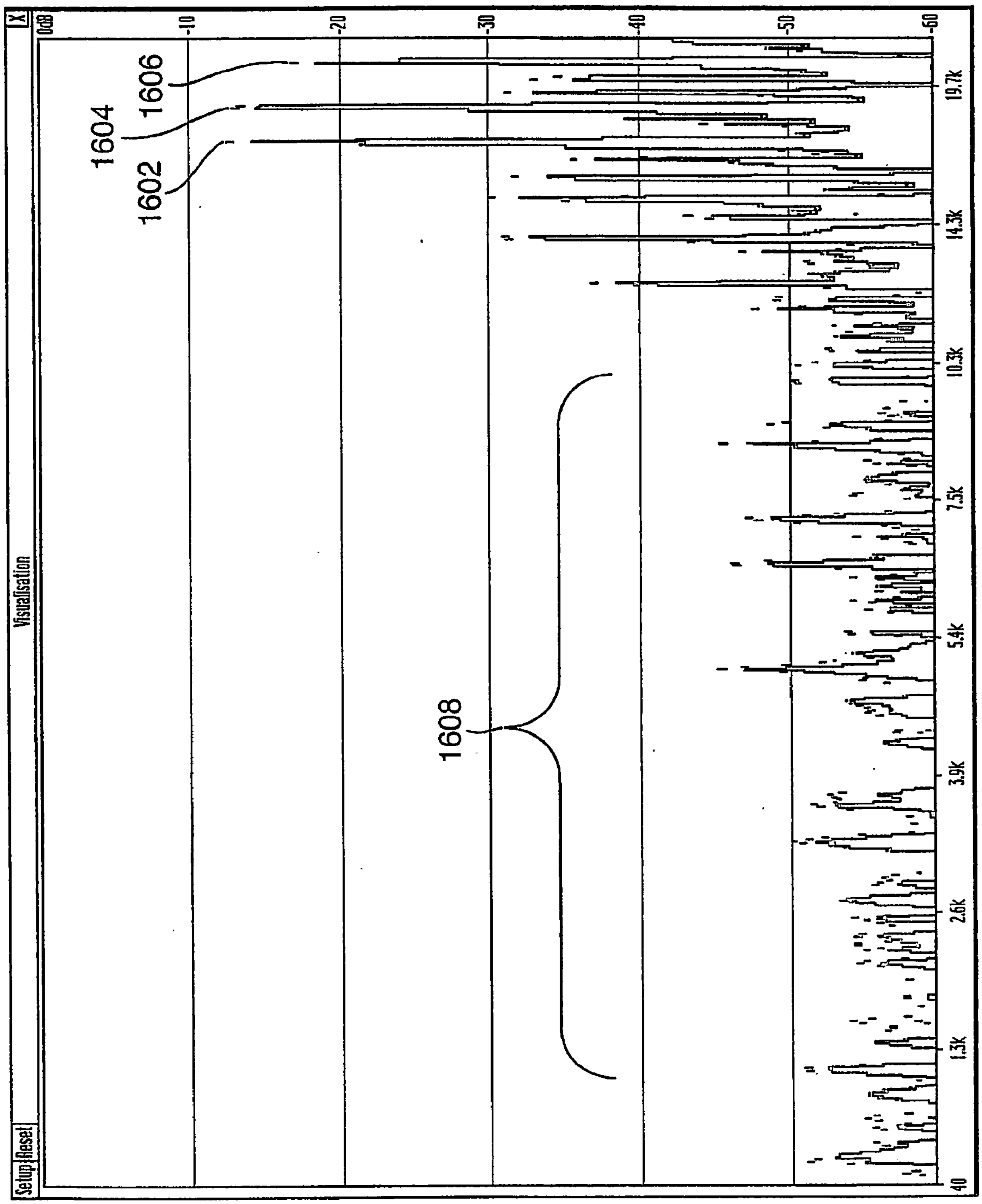
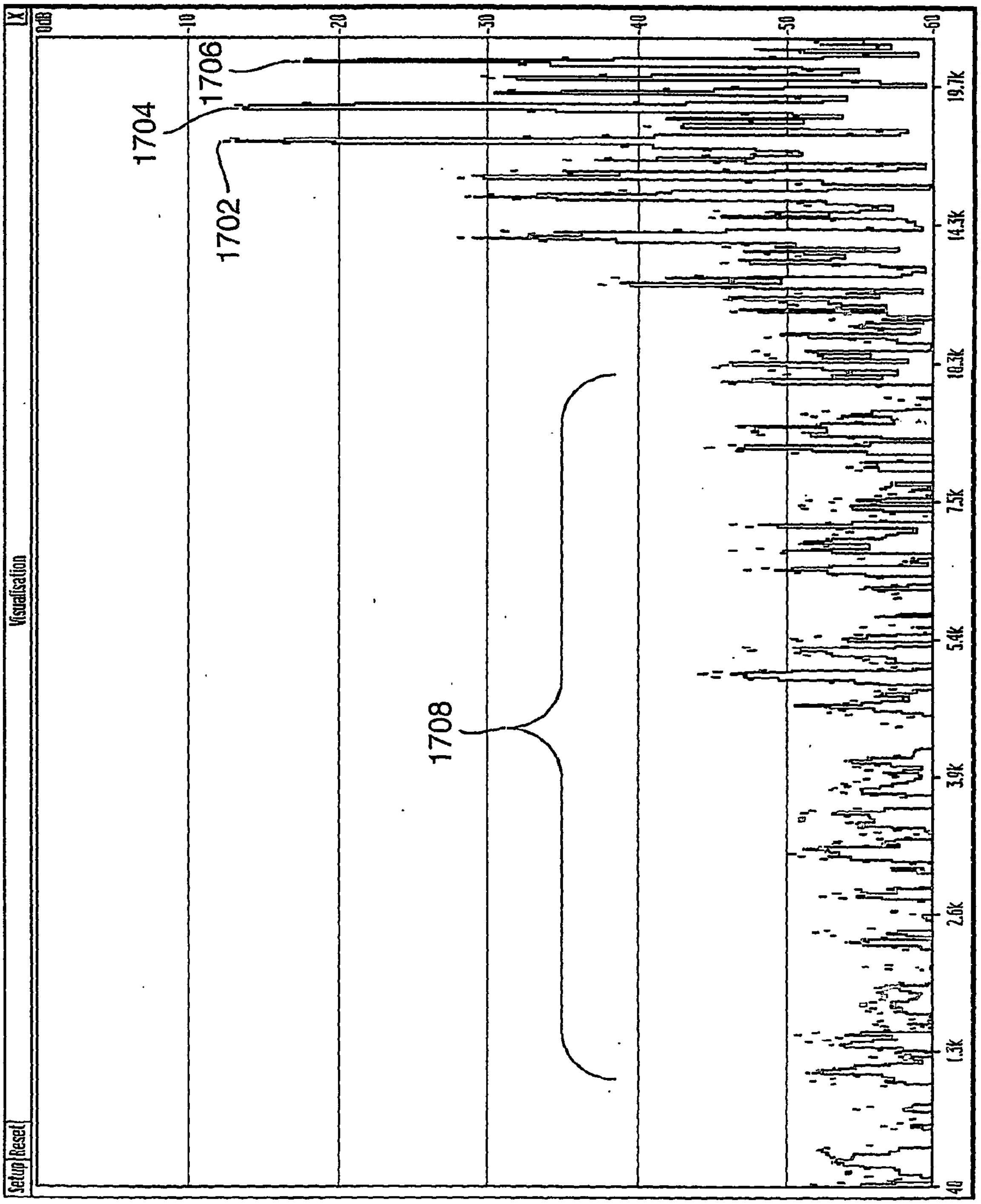

Visualisation
Setup
Reset
X
0dB
-10
-20
-30
-40
-50
-60
40
1.3k
2.6k
3.9k
5.4k
7.5k
10.3k
14.3k
19.7k
1602
1604
1606
1608

Fig.14.


Visualisation
Setup
Reset
X
0dB
-10
-20
-30
-40
-50
-60
40
1.3k
2.6k
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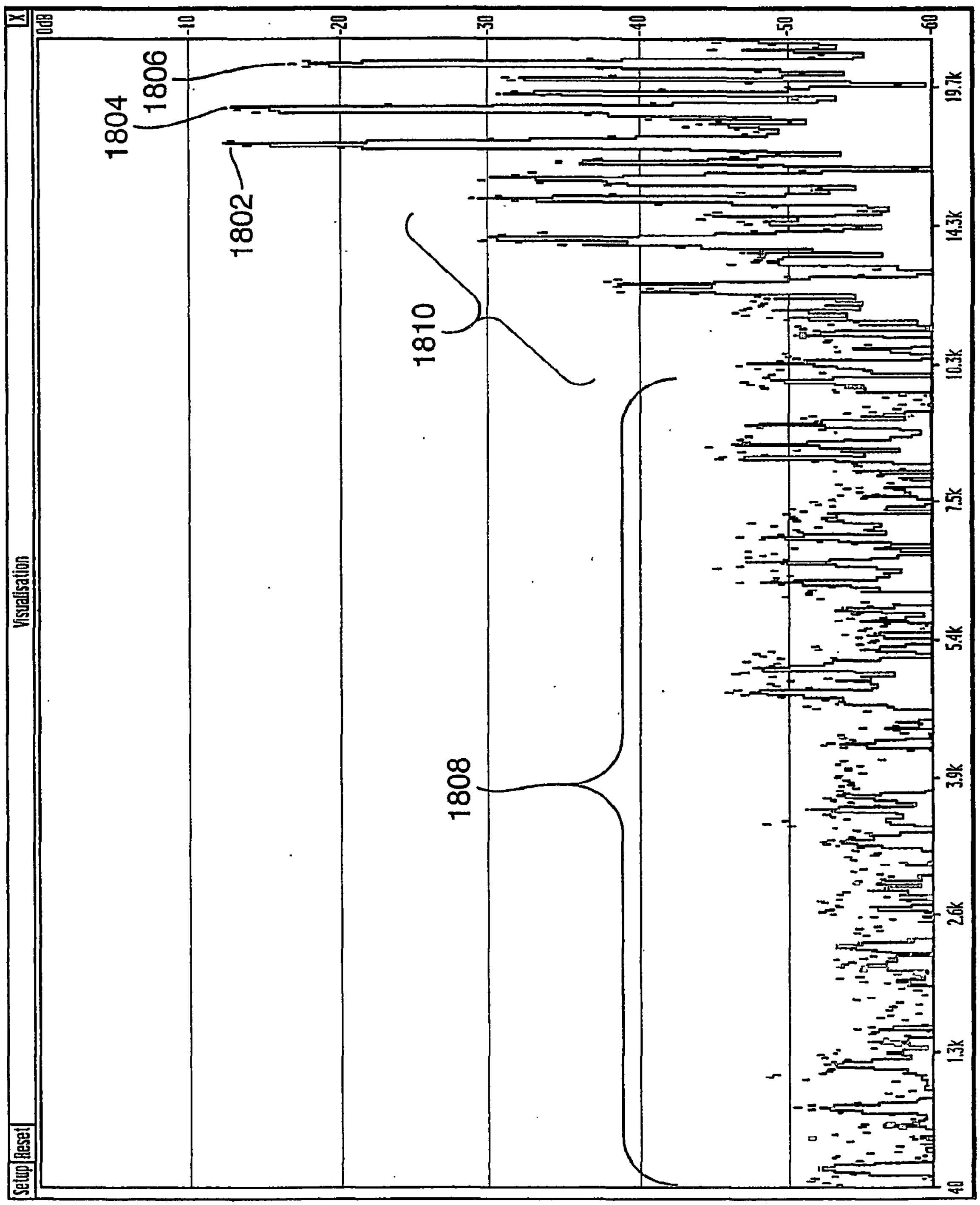
7.5k
10.3k
14.3k
19.7k
1702
1704
1706
1708

Fig.15.


Setup
Reset
Visualisation
X
0dB
-10
-20
-30
-40
-50
-60
40
1.3k
2.6k
3.9k
5.4k
7.5k
10.3k
14.3k
19.7k
1802
1804
1806
1808
1810